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PROVISIONAL APPLICATION FOR PATENT COVER SHEET

This is a request for filing a PROVISIONAL APPLICATION FOR PATENT under 37 CFR 1.53(c).

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☐ Additional inventors are being named on the ____ separately numbered sheets attached hereto

TITLE OF THE INVENTION (280 characters max)

DIGITAL MICROPHONE

CORRESPONDENCE ADDRESS

Direct all correspondence to:

☒ Customer Number: 25269 → Place Customer Number Bar Code Label here

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OR

☐ Firm or Individual Name					
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Respectfully submitted,

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USE ONLY FOR FILING A PROVISIONAL APPLICATION FOR PATENT

INITIAL INFORMATION DATA SHEET

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Digital Microphone

Field of the invention

This invention relates to a digital microphone and an integrated circuit comprising an amplifier with a feedback filter and an analogue-to-digital converter.

Background of the invention

The concept of digital microphones are not new, in fact it is a very obvious idea to combine a microphone preamplifier with an A/D converter. But finding a solution of how to implement a microphone preamplifier and an A/D converter on a single chip with very high performance which can be mass produced at very low cost is not a trivial task. Generally, and also in this specific context, high performance is characterized by low noise, low distortion, large dynamic range and other well known properties.

Expediently, for implementation in consumer/telecommunications products a microphone with such properties should be implemented in a single ASIC (Application Specific Integrated Circuit) with few or no external components and with a minimum die area. Thus achieving low cost, high performance and enabling miniature digital microphones.

Summary of the invention

In the following, firstly, some observations in relation to the present invention is outlined, secondly, objects of the invention are summarised, and thirdly aspects of the invention are summarised.

In recent years so-called sigma delta modulators have become very popular for implementing A/D converters. They exhibit many virtues, which among others are: no need for high precision components; high linearity; and for so-called single loop modulators also the advantages of small die area, low voltage operation and possibly very low power consumption. These are

advantages which makes sigma delta modulators very suitable for single chip implementations.

A special class of sigma delta modulators are 1-bit quantized sigma delta modulators. This type of modulators are especially suited for low cost implementations as the complexity of the analogue part of the A/D converter is minimal compared to other types of A/D converters. A complete 1-bit sigma delta converter consists of a 1-bit analogue sigma delta modulator and a digital decimation filter only. Even the normally required anti aliasing filter can be omitted and replaced by a simple RC-filter. This is due to the fact that heavy over-sampling is used and thus the digital decimation filter performs the job of anti-aliasing filtering.

1-bit sigma delta modulators are very simple to implement in the analogue domain. Thus they are very suitable for low cost miniature digital microphones. Unfortunately they do also have disadvantages. Especially 1bit sigma delta modulators exhibit the so called idle mode tones, which are low level tones in the audio band caused by low frequency or DC levels at the input of the modulator. This is the reason why 1-bit sigma delta modulators has been abandoned by many despite of its many virtues. One can use dither to remove this problem or design chaotic modulators: but all of these solutions has the effect that the complexity of the design increases dramatically. Thus both power consumption and die area increases dramatically.

This 'idle mode tones' effect has caused sigma delta modulators less suitable for high quality audio applications. Apparently, this may seem to be of little concern in consumer/telecom applications. But as the demand for low cost digital microphones increases higher demands of performance, which may almost equal the performance of high quality audio, will follow. Consequently,

the 'idle mode tones' effect will become an increasing problem also for telecom applications.

In order to achieve high performance from the digital microphone, the preamplifier of the digital microphone ASIC has to have as high performance as possible ie low noise, low distortion, high dynamic range etc. According to presently available technology, CMOS technology is a prerequisite to achieve low noise performance and it can be shown that the input stage of the amplifier can be optimized in respect to noise. That is, there exist an area of the input device(s) for which the input related noise is minimum. Also the input impedance should be as large as possible in order to minimize the noise. This is especially dominant for new and thinner types of telecom microphones which has a much lower sensitivity and cartridge capacitance than previously experienced.

Unfortunately this has the consequence that the preamplifier becomes able of amplifying low frequency signals arising from the sound pressure of a door slamming, car rumbling or just changes in sensitivity of the microphone element due to humidity changes. This adds to the above explained problem of idle tone modes if a 1-bit sigma delta modulator is used. In fact also 2-bit and modulators with even more levels will exhibit such behaviour when exposed to such low frequency signals.

Additionally, these low frequency signals reduces the dynamic range and creates inter-modulation distortion as the low frequency signals can be excessive in amplitude.

The problem is worsened as the telecom microphones are becoming smaller and thinner and thus more gain is required from the preamplifier. However, normally the disturbing low frequency signals do not become smaller in amplitude. Thus the relative effect of the disturbance will increase.

So there is a need for a configuration of a preamplifier and an A/D converter which is suited for thin ECM cartridges with a very low cartridge sensitivity and capacitance. Additionally, the configuration should provide a very high performance on noise, dynamic range and distortion. Moreover, it shall be feasible to implement the configuration on a single die with a very small area in combination with few or none external components.

Today the architecture of the mobile phone consists of a two or more separate, integrated circuit chips. Typically, these chips comprise a Digital Signal Processor, DSP, a Radio Frequency, RF, chip and a baseband chip. The baseband chip integrates most low frequency analogue functions. But when the transducers becomes digital and integrates the analogue functionality in the transducer, the baseband chip becomes redundant and a completely new mobile architecture with all analogue functionality build into the transducers will arise. This also means that a regulated, low noise power supply for the analogue circuitry will not be provided in the same way in the future mobile phone architecture. This means that the digital transducer/microphone will have to have this power supple build in.

So there is a need for a digital microphone chip with build in power supply regulator. At the same time this should be implemented with the lowest possible die area and the fewest possible numbers of external components.

The vast majority of microphones for telecom applications has a microphone membrane that is charged with a static electrical charge during manufacturing. If this charge disappears the performance of the microphone is completely or almost completely destroyed. Therefore, these microphones cannot be assembled using reflow soldering since the high temperatures during this process will destroy the permanent charge in the electret and thus d teriorate the performance of the microphone.

Solutions to this problem has been shown, in the form of eg Silicon microphones, so that reflow soldering can be used and thus lower the cost of the assembly process. This solution has the advantage of offering easy assembly of microphones having many more terminals than today's microphones. This is a major advantage in the case of a digital microphone. These solutions require an on-chip voltage pump for biasing the microphone and thus there is a need for a digital chip with added voltage pump and a preamplifier optimized for a silicon microphone. All of this with very low cost added.

To conclude, the topology of the integrated circuit in a digital microphone plays a major role in how the complete digital microphone performs. That is, an optimal topology of the IC will give the best performance vs. cost ratio.

Turning now to different objects of the invention:

It is an objective of the present invention to provide a combination of a preamplifier and a A/D converter with the lowest possible noise, largest dynamic range, low distortion and at the same time having the smallest possible die area.

It is an objective of the present invention to provide a digital amplifier ASIC which is able to handle slowly varying signals with a relatively large amplitude at its input terminal while at the same time being able to amplify a low level signal with a higher frequency without distortion and without creating in-band idle mode tones.

It is an objective of the present invention to provide a digital amplifier ASIC which performance is very insensitive towards leakage and parasitic capacitances connected to the input.

It is an objective of the present invention to provide a single chip solution with all functions integrated. That is a preamplifier, A/D converter, power supply regulator digital processing circuitry etc. All integrated on the same chip with smallest possible die area and few or no external components.

Turning now to different aspects of the invention:

The invention is provided in a **first aspect** by a digital microphone integrated circuit comprising a high-impedant biasing of an amplifier input, an amplifier with a low pass filter feedback, an anti-aliasing filter, and a sigma delta modulator. In accordance with the present invention an expedient architecture of a digital microphone integrated circuit has a preamplifier with a high pass filter function, followed by a low pass filter, followed by a sigma delta modulator or an A/D converter.

This expedient architecture is based on the following signal processing blocks, which are at minimum required in the digital microphone IC: a low pass filter function to filter out the low frequency signals normally present at the input of the preamplifier; a preamplifier to buffer the high impedant microphone element and to amplify the signal in order to obtain a good signal to noise ratio for the overall system; an anti-aliasing filter to filter out the high frequency signal before they are sampled by the A/D converter, which can be implemented as a sigma delta converter.

As cost is a major concern in telecom microphones the topology should have the lowest possible cost for a given performance or the best given performance for a given cost. For analogue signal processing current consumption, signal to noise ratio, area and signal swings are dependent and highly correlated parameters. That is, on the one hand, in order to get a high signal to noise ratio, large area and current are needed. On the other hand,

however, if the signal wing is large the area savings and or current savings can be obtained. These observations can be used to optimize the system/topology for optimum performance or lowest cost.

Each of the above mentioned blocks can be analysed. First we will look at the two analogue filters i.e. the high pass filter and the low pass filter. The simplest forms of these consists of one resistor and one capacitor. Here the interesting parameters are area and noise. I.e. how should the capacitance and resistor values be chosen for a given signal to noise ratio in order to obtain the smallest possible area? For the high pass filter the noise from the resistor is low pass filtered by the coupling capacitance which means if the cut-off of the high pass filter is located at a low frequency e.g. Below the cut-off of the A-weighting function then the noise of the high pass filter can be minimized. As one normally is interested in having the cut-off of the high pass filter below the cut off of the A-weighting e.g. 100-200Hz, the size of the high pass filter can be made relatively small. The low pass filter (anti-aliasing filter) is different here since the noise can be minimized by moving the cut-off to a high frequency. The function of the low pass filter is that of an anti-aliasing filter and thus it should be have a cut-off frequency as low as possible. This is in contrast of what is needed to optimize for lowest area for a given signal to noise ratio. So it can be concluded that the low pass filter is more critical regarding noise and area than the high pass filter.

In order to optimize the signal to noise ratio in a chain of signal processing blocks the block closest to the source should have the largest gain. I.e. the preamplifier should have a large gain.

We can now describe how the optimal architecture when optimizing for signal to noise ratio and or area will look like. In order to obtain the best signal to noise ratio for a capacitive microphone the input bias device of the preamplifier should have an impedance as large as possible. Thus pushing

the noise from the input bias device to a frequency so low that the A-weighing weighs it out.

This will give the lowest possible noise but also very low settling of the input signal after power on of the system. Also slowly varying signals appearing from change in cartridge sensitivity due to humidity changes etc. will not be attenuated due to the low frequency cut-off of the biasing device.

The preamplifier must be connected directly to the microphone cartridge. And have a large gain in order to optimize overall noise performance. In order not to overload the preamp and to avoid that low frequency signal reaches the input of the modulator where in can cause the modulator to create idle mode tones. From this perspective the high pass filter should be placed as close to the preamp as possible. But obviously it is not possible to put it at the input as it would load the cartridge and putting it after the amplifier will not assure that the preamp doesn't gets overloaded. The solution is to integrate the filter with the preamp ie to make a preamp with a high pass filter function and gain.

The low pass filter can now be put directly after the preamplifier as it should remove any high frequency signal before the signal is sampled ie preventing high frequency signals entering the sigma delta modulator. Because of the high gain of the preamplifier the noise of the low pass filter is now much less important and thus the area of this filter can be greatly reduced.

So to conclude, a preferred microphone circuit architecture has a preamplifier with a high pass filter function, followed by as low pass filter, followed by a sigma delta modulator or an A/D converter.

This architecture is the optimum in respect of achieving the best performance and at the same time smallest die area. The optimum architecture is

independent on how the different blocks are implemented. I.e. The preamplifier can be implemented as differential as well as single ended. Any of the blocks can be made single ended or differential. Or any combination hereoff. Or extra signal processing blocks such as filters buffers or amplifiers can be added. The arguments for the optimal achitecture will still hold.

The invention is provided in **second aspect** by using a preamplifier having a lower gain at low frequencies compared to audio frequencies.
In order not to overload the preamplifier with low frequency signals then the gain at low frequencies should be low. This also goes for a differential implementation. But in this case the it is the single ended gain which should be lower at low frequencies than at audio frequencies.

The invention is provided in a **third aspect** by using a differential preamplifier converting the input signal into a common mode signal for low frequencies and into a differential signal for audio frequencies.

In a preferred embodiment, the differential preamplifier is configured as an instrumentation type amplifier with two inputs and a first and a second output, wherein the first and second input is arranged to receive a microphone signal, but wherein the inputs are coupled to receive the microphone signals substantially in phase at relatively low frequencies and substantially out of phase at relatively high frequencies. Preferably, the relative high frequencies comprises the audible frequency range and higher frequencies.

The second input can be coupled to receive the microphone signal from the first input through a phase shifter, which has a substantial constant phase shift in the audible range and has a gradually shifting phase for gradually lower frequencies until a phase shift of about 180 +/- 5, 10, 15, or 20 degrees relative to the phase shift in the audible range is realised. At that frequency the phase shift remains substantial constant for lower frequencies. The

phase shifter can be of any order eg 1st order, 2nd order, 3rd order, 4th order or any higher order. However, a 1st order is preferred with regard to realise low chip die and low cost.

Alternatively, the second input can be coupled to receive the microphone signal from the first output through the phase shifter.

Preferably, the phase shift and the filter function is realised in an instrumentation amplifier by means of a RC circuit comprising a first and a second feedback path, each path coupled between the output and input of a respective amplifier in the instrumentation amplifier, and each path comprising a parallel coupling of a resistor and a capacitor, and both paths being coupled together at the amplifier input side of the paths by means of a capacitor.

The differential preamplifier is configured to provide frequencies below an audio band as common mode signals and audio band signals as differential mode signals. A circuit coupled to receive the differential signal will be able to suppress common mode signals and thus low frequency signals far better than if ordinary high pass filtering was used. Despite the increased complexity of using differential mode circuits as compared to single ended circuits the ability of suppressing low frequent signals has proved to be very effective – also from a cost versus performance perspective. The frequency dependent phase shift required to achieve the common mode operation at low frequencies and the differential mode operation at higher frequencies can be obtained in various ways. Three types of achieving this phase shift are:

- A phase shifter coupled between inputs of the differential amplifier;
- A phase shifter cross coupled between an output of one side of the differential amplifier and an input of the opposite side of the differential amplifier;
- A combination of the above two types;

- A phase shifter cross coupled between a signal node at which a signal substantially in phase with the input signal to the preamplifier is present and an input of the opposite side of the differential amplifier

The invention is provide in a **fourth aspect** by using a simple 1st order RC filter as an anti-aliasing filter in combination with a high over sampling factor.

In order to obtain the best signal to noise and at the same time a low current consumption then the anti-aliasing (AA) filter has to be as simple as possible. The simplest possible filer is a simple 1st order RC filter. Any active component will add to the current consumption and add to the noise. A simple first. E.g. A simple RC filter will have the lowest noise and the lowest power consumption. In order achieve sufficient AA- filtering the ratio between the AA filter cut-off and the sampling frequency has to be sufficiently large. As explained in the section on the optimal architecture there are natural limits in how low the AA filter frequency can be made in order to achieve low noise and small die area. This has the consequence that the sampling frequency should be relatively high. The current consumption of the A/D converter will be marginally higher with the very high sampling frequency but all in alll the solution with a very simple RC filter as AA filter proves the best overall results in respect to low noise, small die area and low current consumption. The preferred implementation is a simple RC filter with no buffer or amplifier between the modulator and the filter. As this would only reduce the signal to noise ratio, increase the current consumption and the die area.

The invention is provide in a **fifth aspect** by using an over-sampled sigma delta converter. Apart from the advantage of using a over sampled AD-Converter in conjunction with a simple RC filter using over sampled A/D converters also has advantage in terms of smaller die area. I.e. For a given noise and current consumption specification the size of the capacitors becomes smaller the larger the over sampling factor. This is under the

assumption that the technology (E.g. CMOS) doesn't limits the performance first. E.g. For a 0.5um CMOS the capacitors can be scaled with very small penalties upto a clock frequency of 4-6 MHz. Which for audio purpose AD converters gives a large over sampling factor. For more modern technology this limit are moved even further up.

The invention is provide in a **sixth aspect** by integrating a voltage regulator on chip the same chip as the amplifier and the A/D-converter.

By implementing the regulator on chip possible savings in other parts of the system are possible. I.e. A possible all digital mobile terminal architecture. Furthermore by adding the voltage regulator on chip the possibility of design a voltage regulator that uses no external components Exists. This is a very effective way of reducing cost as the obviously the cost of an external component has been removed. But also the need for an extra pad for the external component has been removed and thus making the die area smaller. The only way of designing a cost effective on chip regulator without the need of an external component is to use a regulator with a source follower output. Unfortunately this normally has a large drop out voltage. I.e. Larger than the gate source voltage. This can though be overcome by using a switched circuitry to increase the gate voltage level to above the supply voltage. Thus also allowing low voltage operation with a source follower regulator.

The invention is provide in a **seventh aspect** by integrating digital signal processing and or digital functionality on the same chip as the amplifier and the A/D-converter. Cost savings can be achieved in many ways, some on component level others on architectural level. E.g. the architecture of a mobil phone. Some parts of the digital signal processing are best done close to the microphone and can thus enable cost savings on the architectual level.

The invention is provided in a **eight aspect** by integrating a voltage pump on the same chip as the amplifier and the A/D-converter. Some microphone types requires a bias voltage in order to function. E.g. Silicon microphones. There are several advantage of integrating this bias voltage on the same chip as the preamp and the A/D converter.

Such as bias voltage is normally higher than the supply voltage. In fact in can be as high as 30V and thus many times higher than the power supply. Such a bias voltage is generated using a voltage pump which typically can consist of a Dickson pump, an oscillator and some kind of reference.
In the case that the pump is integrated together with the A/D, a clock frequency is already present and the oscillator is not needed. Furthermore as the clock for the A/D and for the voltage pump the clock noise from the voltage pump is synchronised with the sampling and is thus a less severe problem.

If the microphone is biased at a DC high voltage then a DC coupling capacitance is needed as then amplifier in nearly all cases are not able to handle the large DC level without overload. Furthermore by integrating everything on the same chip the total performance can be optimized giving the best possible performance.

The invention is provided in a **ninth aspect** by using a preamplifier with build in offset.

In all the mentioned amplifier configurations it is not shown how the DC bias points are set. Especially in the case of a single ended power supply the setting of the bias points are an important detail. E.g. The outputs of the differential amplifier should idealy have a DC level equal to half of the power supply for the differential preamplifier. I.e. In order to allow the largest

possible signal swing at the output. One way is to build in an offset in the preamplifier. In case of a differential preamp it can be build into one of the preamps or both. Another solution is to apply a offset at one of the inputs of the amplifier. In that case a DC coupling is needed to separate the DC on the input from the microphone.

Brief description of the drawings

The invention will be described in more detail and with reference to a preferred embodiment, in which:

fig. 1 shows a digital microphone comprising an electret microphone member, a differential amplifier and an A/D converter;

fig. 2 shows a differential amplifier with input and output terminals and signals illustrating a low frequency behaviour of the differential amplifier;

fig. 3 shows the differential amplifier with input and output terminals and signals illustrating a high frequency behaviour of the differential amplifier;

fig. 4 shows a portion of a digital microphone comprising an electret microphone member and a differential amplifier in a first configuration;

fig. 5 shows a portion of a digital microphone comprising an electret microphone member and a differential amplifier in a second configuration;

fig. 6 shows a portion of a digital microphone comprising an electret microphone member and a differential amplifier with a feedback filter;

fig. 7 shows a preferred embodiment of a feedback filter;

fig. 8 shows a switch-capacitor detection of a differential signal to be input to an A/D converter;

fig. 9 shows a first digital microphone;

fig. 10 shows a second digital microphone;

fig. 11 is a block diagram of a digital microphone;

fig. 12 is another block diagram of a digital microphone; and

fig. 13 is a schematic view of a microphone with an integrated circuit and a microphone member.

Fig. 1 shows a digital microphone comprising an electret microphone member, a differential amplifier and an A/D converter. The electret microphone member is biased via a bias resistor 104 that is coupled to a bias voltage Vb. Thereby an electric charge is provided to the membrane or movable member of the microphone 105, Cmic. A signal provided in response to a sound pressure on the microphone and thus making the membrane move is provided to an amplifier 101. The amplifier 101 is characterised by having a gain characteristic with relative low gain for frequencies below an audible range and a relative high gain for frequencies in the audible range. Preferably, the gain characteristic descents as a 1st, 2nd, 3rd, 4th, or higher order below the audible range. In addition thereto the amplifier is characterised by processing a low frequency microphone signal as a common-mode signal and a high frequency microphone signal as a differential mode signal. Thereby low frequency components are effectively suppressed.

The output of the amplifier 101 is provided to a sigma-delta A/D converter 103 via an anti-aliasing filter, AAF, 102. The sigma-delta converter provides an over-sampled 1-bit output signal.

Fig. 2 shows a differential amplifier with input and output terminals and signals illustrating a low frequency behaviour of the differential amplifier. The signal processing of amplifier 101 at low frequencies is illustrated. The curve 201 illustrates a microphone signal in the time-domain input to the amplifier (ϕ) and at a low frequency. The curves 202 and 203 illustrates that respective outputs (φ, φ^*) of the amplifier are substantial in phase and thus represent a common-mode differential signal.

Fig. 3 shows the differential amplifier with input and output terminals and signals illustrating a high frequency behaviour of the differential amplifier. The

signal processing of amplifier 101 at high, audio band, frequencies is illustrated. The curve 301 illustrates a microphone signal in the time-domain input to the amplifier (ϕ) and at a audio frequency. The curves 302 and 303 illustrates that respective outputs (φ, φ*) of the amplifier are substantial 180 degrees out of phase and thus represent a differential-mode differential signal.

Fig. 4 shows a portion of a digital microphone comprising an electret microphone member and a differential amplifier in a first configuration.

Fig. 5 shows a portion of a digital microphone comprising an electret microphone member and a differential amplifier in a second configuration. The electret microphone member is biased via a bias resistor 504 that is coupled to a bias voltage Vb. Thereby an electric charge is provided to the membrane or movable member of the microphone 505, Cmic. In order to block the DC bias voltage, Vb, from the input of the differential amplifier 510 a capacitor 506 is applied. The differential amplifier 510 is configured as a so-called instrumentation amplifier wherein two operation amplifiers 501 and 502 each are coupled with a feedback path from their respective outputs φ, φ* to their respective inverting input terminal. The inverting inputs of the operation amplifiers are coupled together by means of a capacitor 508. A non-inverting input of the one operational amplifier 501 is coupled to receive the microphone signal via the DC-blocking capacitor 506. A non-inverting input of the other operational amplifier 502 is coupled to receive a feedback signal from the output φ of the other operational amplifier via a resistor 512. The non-inverting input is also coupled to ground by means of a capacitor 513.

The feedback path of the operational amplifier 501 comprises a resistor 503 and a capacitor 509 coupled in parallel to constitute a first order filter. Likewise, the feedback path of the operational amplifier 502 comprises a

resistor 511 and a capacitor 507 coupled in parallel to constitute a first order filter.

The RC network comprising resistor 512 and capacitor 513 is configured to provide a gradually shifting phase of the signal

The phase shift between the one side, constituted around operational amplifier 501, of the differential amplifier and the other side, constituted around operational amplifier 502 is implemented partly by capacitor 508 and partly by the RC filter 512, 513. Thus, the phase shift is obtained by a phase shifter, capacitor 508, coupled between inputs of the differential amplifier and a phase shifter, capacitor 513 and resistor 512, cross coupled between an output of one side of the differential amplifier and an input of the opposite side of the differential amplifier. Thus, the effective phase shift is obtained by means of two phase shifters. However, one of such two coupled phase shifters may be sufficient to establish the effective phase shift. Likewise, other configurations of phase shifters can be embodied without departing from the scope of the invention.

Fig. 6 shows a portion of a digital microphone comprising an electret microphone member and a differential amplifier with a feedback filter. In this illustration a differential amplifier 607 with a first and a second operational amplifier is shown with a filter block 603. The filter block 603 implements feedback paths of the respective operational amplifiers and coupling of the inverting inputs of the respective operational amplifiers 601 and 602.

The filter block can implement a filter with two feedback paths of any order eg a 1st order, 2nd order, 3rd order, 4th order or any higher order.

Fig. 7 shows a preferred embodiment of a feedback filter. The feedback filter can implement the filter block 603 of fig. 6.

Fig. 8 shows a switch-capacitor detection of a differential signal to be input to an A/D converter. A switch-capacitor detection of differential signals is known to a person skilled in the art, but is shown to illustrate interconnection of a differential amplifier with a below audio band cut-off characteristic and a sigma-delta converter. In this illustration an anti-aliasing filter is not shown, but would typically be required to remove spectral components above a sampling frequency of the converter.

Fig. 9 shows a first digital microphone. In this aspect of the invention, the digital microphone 901 comprises an integrated circuit 902 with a microphone voltage bias circuit 903, 904; an amplifier 905 with a transfer characteristic which suppresses spectral components below an audio band and provides a substantial flat frequency response for audio frequencies at a nominal gain; an anti-aliasing filter 906 and a sigma-delta converter 907. The integrated circuit 902 comprises terminals Tc1, Tc2, Tc3, Tc4 for coupling to the microphone element 908, the bias voltage, a ground reference potential and a supply voltage, respectively. Terminal Tc6 provides a digital signal from the A/D converter and via terminal Tc5 a clock signal is provided to the A/D converter. The supply voltage to the amplifier 905 and the sigma-delta converter can be provided via the terminal Tc6, in which case the terminal Tc4 can be omitted.

Fig. 10 shows a second digital microphone. In this aspect of the invention, the digital microphone 1001 comprises an integrated circuit 1002 with a DC voltage regulator 1003 which provides a regulated voltage to the amplifier 1009 and the sigma-delta converter 1011. The microphone bias voltage is provided by an on-chip voltage up-converter 1004 which receives an off-chip oscillating signal with a voltage amplitude; in response thereto the up-converter provides an output oscillating signal with a larger voltage amplitude. This output signal is low pass filtered by low pass filter 1005 and

provided via a series resistor 1006 to the microphone lement 1008. A capacitor 1007 blocks the DC bias voltage from the input of the amplifier 1009 with the transfer function mentioned above. Output of the amplifier 1009 is provided to an anti-aliasing filter 1010 before being input to a sigma-delta converter 1011.

The voltage up-converter or voltage pump, UPC, 1004 can be in the form of a so-called Dickson-converter. The voltage pump is operated by an oscillator which preferably provides a square-wave oscillator signal to the voltage pump. Other signals, eg sine waves or filtered square waves, with lower contents of harmonics may be used to obtain lower noise. As an alternative, the oscillator can be embedded on the chip 1002.

It is shown that the up-converter and the sigma delta converter shares the same oscillator/clock signal as provided via terminal Tc4. It should be noted that the signal may be divided to obtain different oscillating/clock signal frequencies to the UPC and sigma-delta converter.

Fig. 11 is a block diagram of a digital microphone.

Fig. 12 is another block diagram of a digital microphone. Here, a buffer or buffer amplifier is inserted before the sigma-delta converter.

Fig. 13 is a schematic view of a microphone with an integrated circuit and a microphone member. The microphone is shown as a cartridge with a microphone member, comprising the microphone membrane and an integrated circuit.

Generally, it should be noted that the amplifier of fig. 2, 3, 4, 5 and 6 can be applied in an analogue microphone where a pre-amplified microphone signal is provided as an analogue signal. The analogue signal can be provided as a

single ended signal or a differential signal depending on the configuration of the amplifier.

Embodiments with a single-ended configuration

A complete single ended solution is implemented as described in the following. First a single ended preamplifier with a high pass filter function. Then followed by a single ended AA filter. And lastly a single ended modulator/A/D converter. By a single ended A/D converter or modulator is meant a modulator or A/D converter which processes the signal single ended. In nearly all cases the modulator or A/D converter will have two inputs. E.g. a switched capacitor modulator or A/D converter will have an input capacitor which is switched from one node to another going from the high phase to the low phase of the clock. A switched capacitor circuit works by measuring charging and discharging of capacitors. So the input capacitor will have to be switched between two nodes. I.e. effectively the modulator will always have two inputs. As a single ended AA filter is used which only has one output then the other input has to be connected to a reference voltage. This should preferably be equal to the DC level of the output of the AA filter. This reference voltage will always add noise but no signal. Compared to a differential solution which also adds signal then signal to noise ration will thus be lower for the single ended version. It should be noted that the A/D converter can be implemented using other techniques than switched capacitor without departing from the scope of the invention.

Effectively it means that an implementation using a fully differential preamplifier and AA filter will have better performance than a fully single ended version. The modulator can be both single ended and differential. A single ended version of the modulator or the A/D converter will though typically have a small die area.

Cut-off frequencies

The transfer function of the differential amplifier from single ended input to differential output will as explained, at low frequencies be a common mode signal. I.e. the transfer function will be zero and for higher frequencies it will be a differential signal. Looking at the transfer function to differential output it will typically look as a 1st order high pass filter with a cut-off frequency around 1-200Hz typically at 10Hz.

The transfer function from the input to each of the outputs looking on them as single ended outputs will typically also be a 1st order high pass filter. Normally one would want to position this below the cut-of of the differential transfer function. As this gives lower noise. This would normally, because of low frequency signals, overload subsequent stages. E.g. AA filter + Modulator. But as the following blocks can be designed to have a large common mode input range and are only detecting the differential signal it is possible to design a preamplifier which both suppresses the low frequency signals and still have good low noise performance.

Microphones elements

In this invention a grounded microphone element has been shown as this is most common. But floating microphone elements also exists and here the advantages of the differential preamplifier becomes even more pronounced.

In preferred embodiments, the microphone element and the integrated circuit is implemented on a single chip.

The above features may be applied in embodiments of a preamplifier configuration that comprises a gain stage with a feedback filter, where the configuration has a relatively low gain response for frequencies below an audio band and has a relatively high and substantially flat gain response in the audio band. The audio band can be defined to be any band within the typical definition of an audio band. A typical definition can be 20Hz to 20KHz.

Exemplary lower cut-off frequencies for an audio band can be: 20 Hz, 50Hz, 80 Hz, 100Hz, 150 Hz, 200Hz, 250hz. Exemplary upper cut-off frequencies the an audio band could be 3KHz, 5KHz, 8KHz, 10KHz, 18KHz, 20KHz. By substantial flat is meant gain response variations within approximately +/-1dB; +/-3dB; +/-4dB; +/-6dB. However, other additional values of variation can be used to define the term 'substantial flat'. In this case the bands are defined for audio use but the same arguments can be used for other frequency bands. E.g. Ultrasound frequencies in the range from 100kHz to 20MHz.

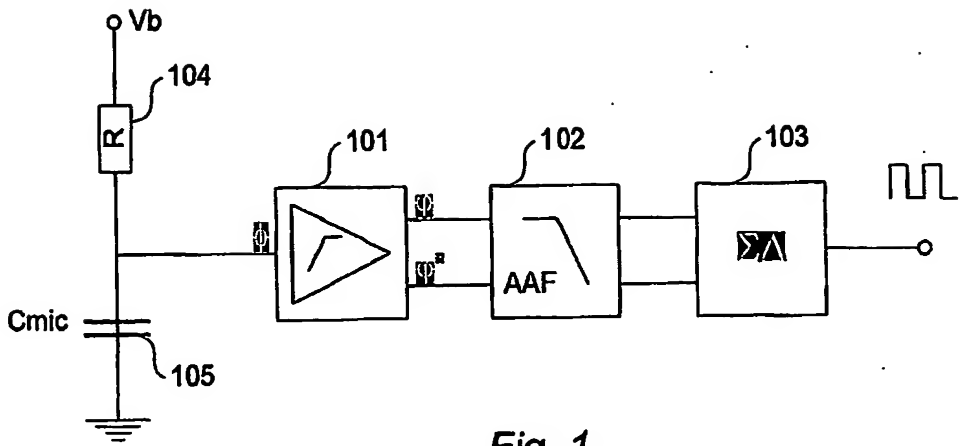


Fig. 1

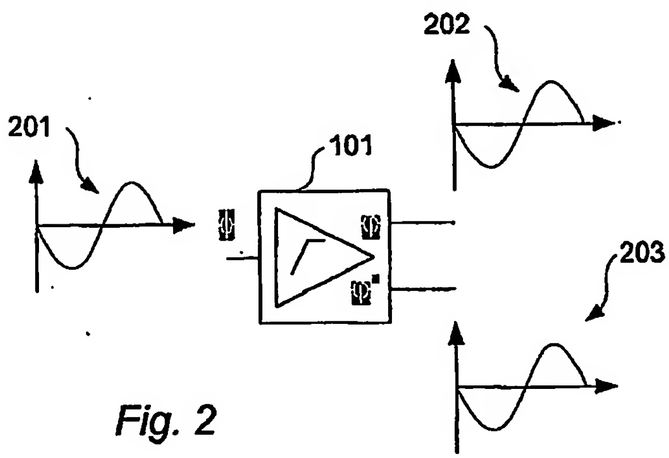


Fig. 2

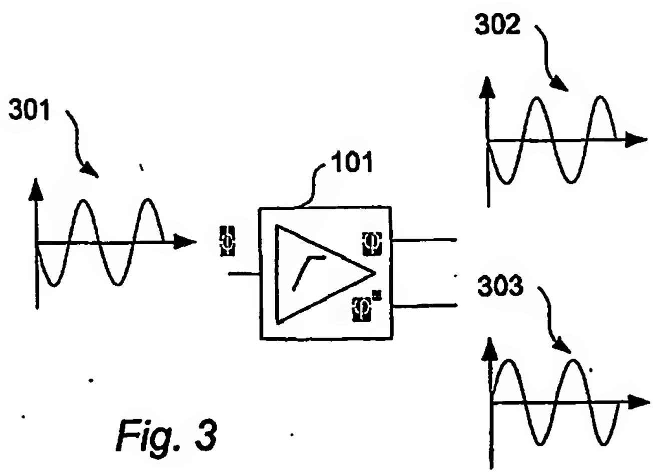


Fig. 3

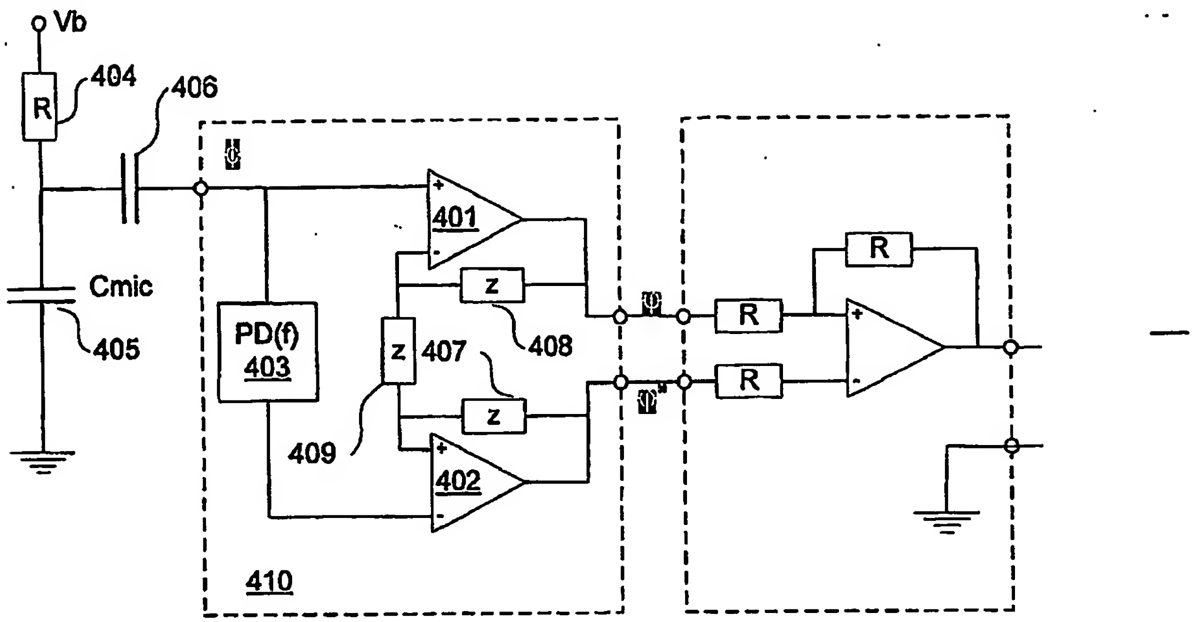
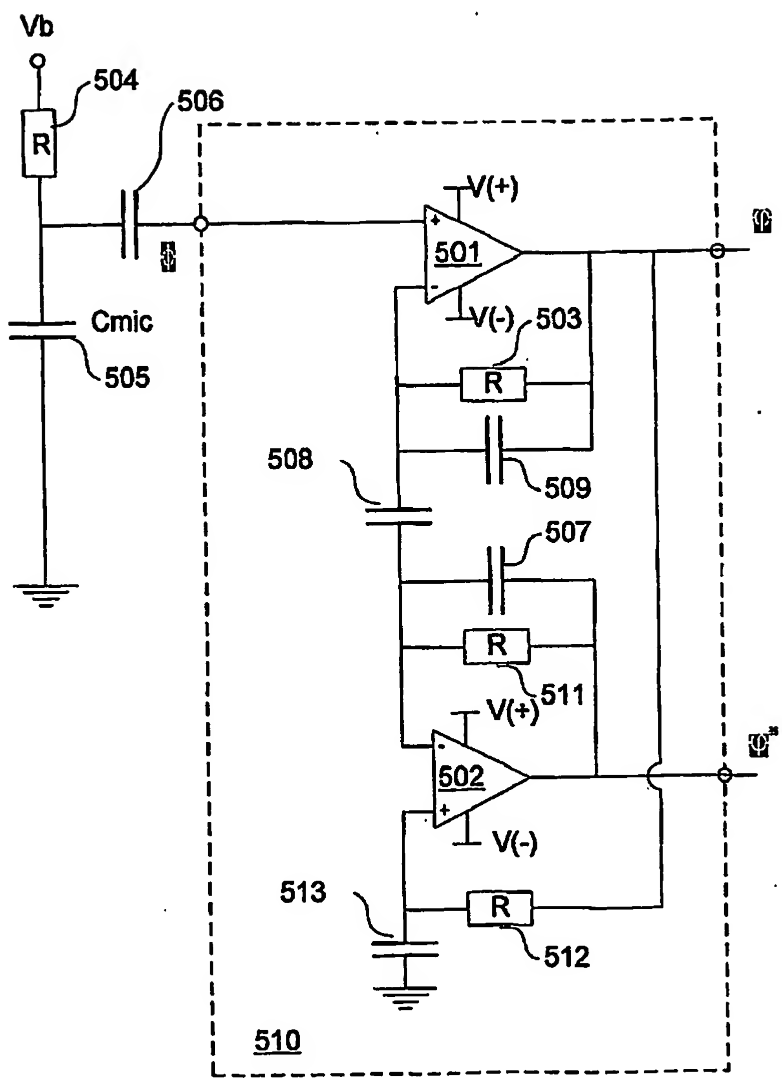


Fig. 4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R
506
Cmic
505
V(+)
501
V(-)
503
R
508
509
507
R
511
V(+)
502
V(-)
513
R
512
510

Fig. 5

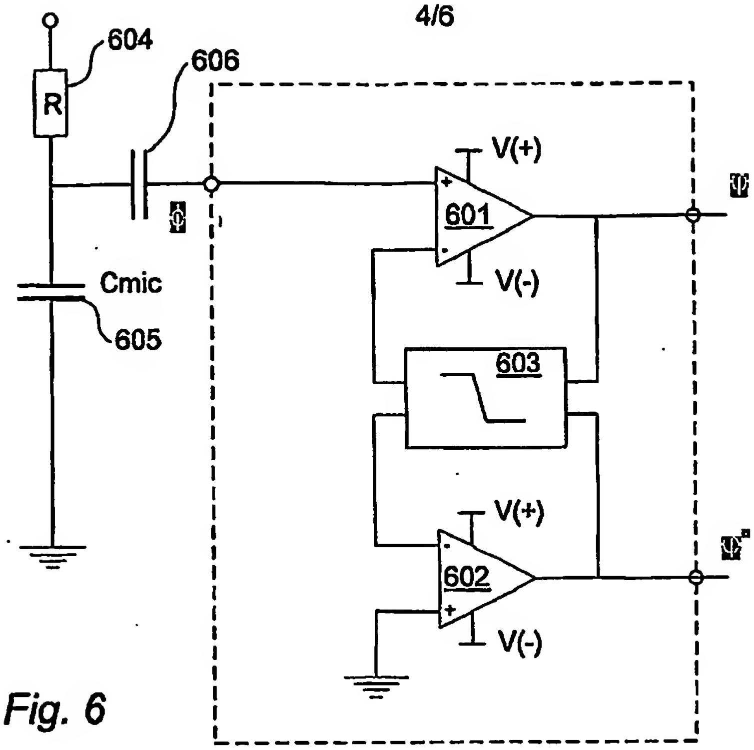


Fig. 6

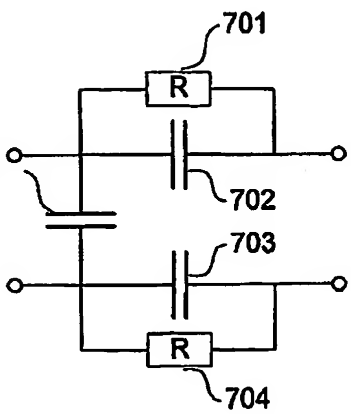


Fig. 7

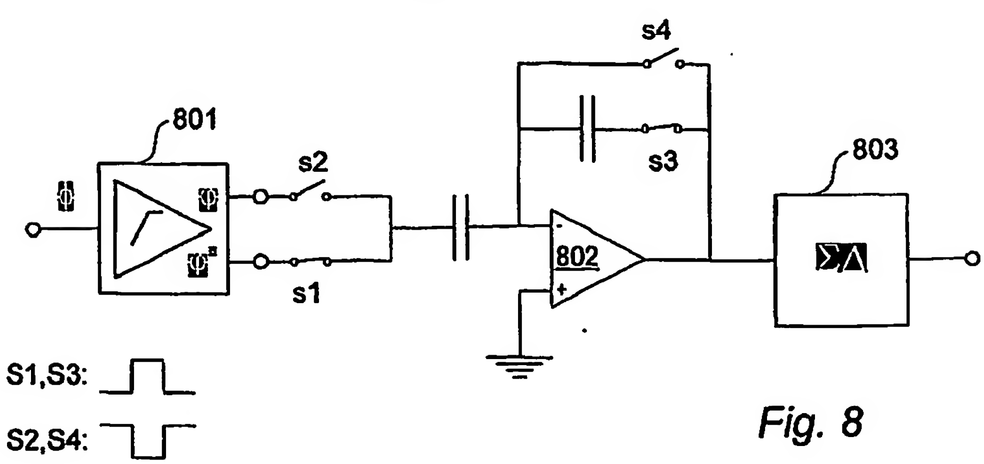
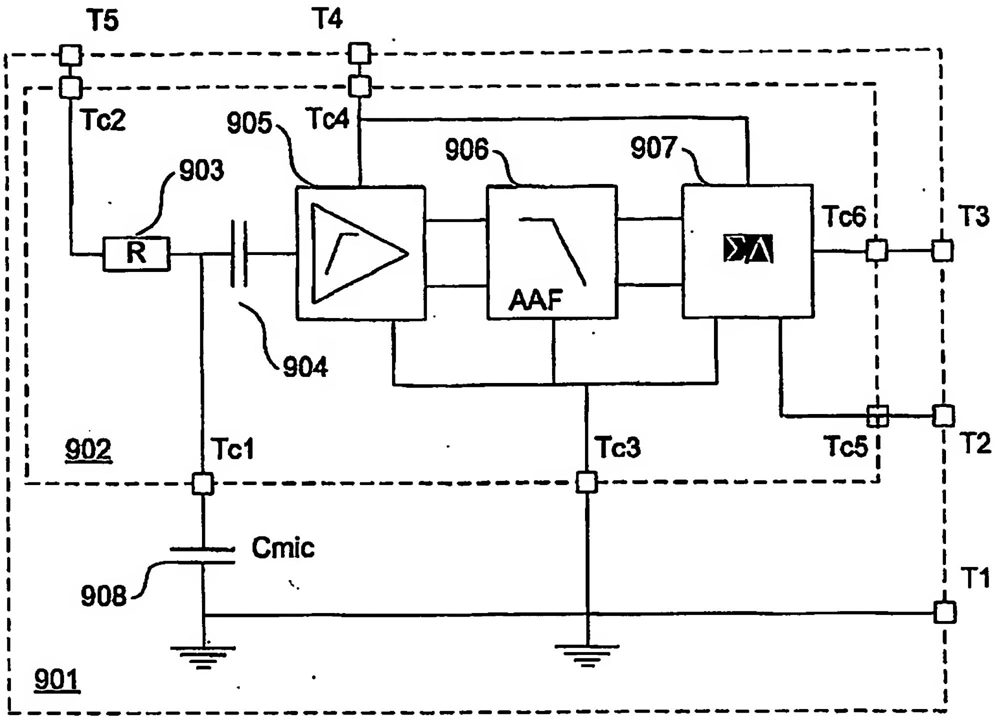


Fig. 8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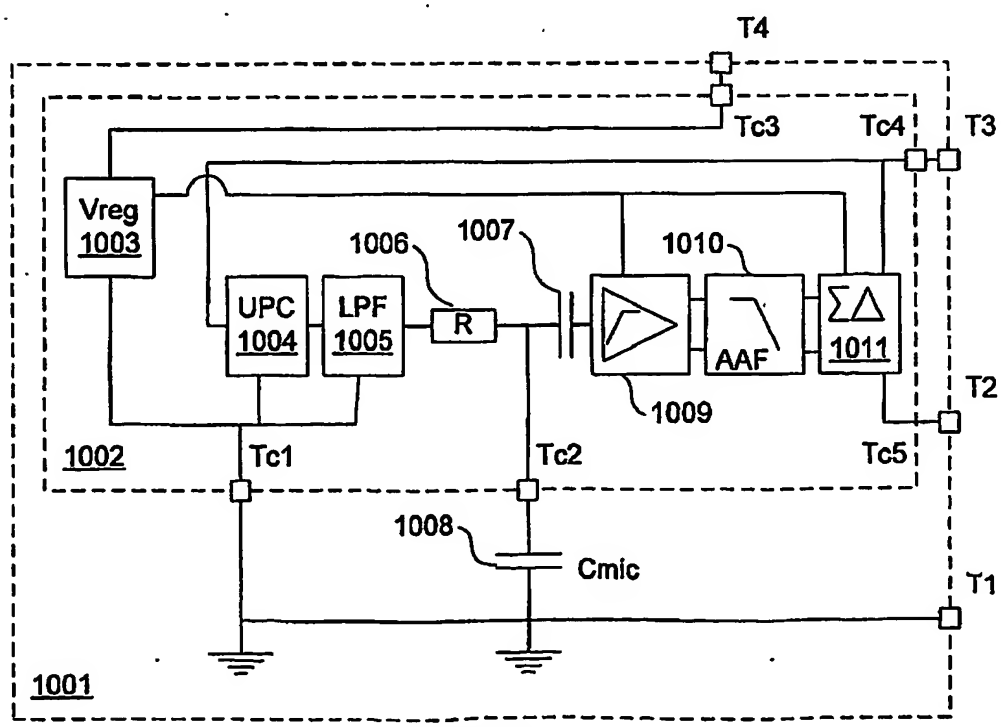
905
Tc4
906
907
903
R
AAF
ΣΔ
Tc6
T3
904
902
Tc1
Tc3
Tc5
T2
Cmic
908
T1
901

Fig. 9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Tc3
Tc4
T3
Vreg
1003
1006
1007
1010
UPC
1004
LPF
1005
R
AAF
ΣΔ
1011
1009
T2
1002
Tc1
Tc2
Tc5
1008
Cmic
T1
1001

Fig. 10

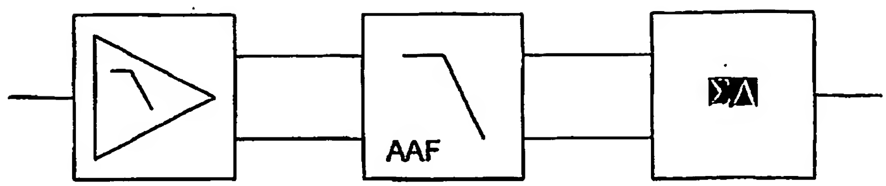

Fig. 11

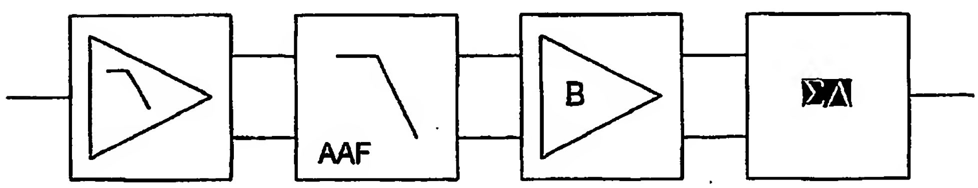

Fig. 12

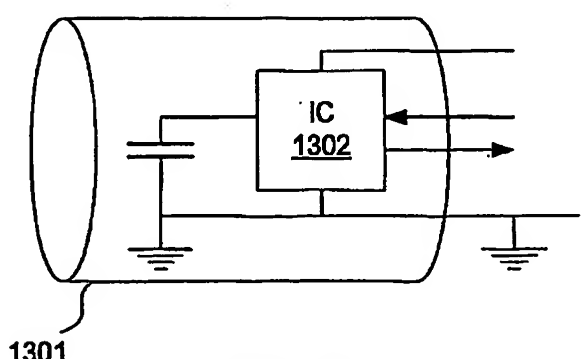

Fig. 13